RESEARCH DEPARTMENT

The Television Centre ambiophony system

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THE BRITISH BROADCASTING CORPORATION
ENGINEERING DIVISION

RESEARCH DEPARTMENT

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SUMMARY

This report gives the results of an investigation of the ambiophony system which has been installed in Television Centre Studio 4 for some years. For the purpose of the investigation, the equipment bay and 30 loudspeakers were transferred to Kingswood Warren Large Viewing Room where uninterrupted work could be carried out.

1. INTRODUCTION

The ambiophony system which has been installed for the past few years in Television Centre Studio 4 follows the design described by Vermeulen.[1] The object of the equipment is to provide artificial reverberation in the studio itself instead of adding it to the microphone signal by means of a separate echo room or reverberation plate.

Sound in the studio is received by a microphone connected through amplifiers to a recording head in a magnetic tape system. The tape is in the form of a loop which passes also over eight reproducing heads, the output of any one of which can be fed back to the recording head. Groups of loudspeakers distributed on the walls and ceiling of the studio are fed from the reproducing heads. The sound issuing from each loudspeaker is delayed in relation to the direct sound received by the ambiophony microphone by an interval equal to the time taken for an element of the tape to travel from the recording head to the relevant reproducing head. Means are provided for adjusting the delay of each head in steps of 15 ms but adjacent heads cannot have relative delays of less than 30 ms. The gain in the internal feedback chain can be varied to give different reverberation times and the same effect can be produced by altering the gains of the loudspeaker amplifiers.

In any system of this kind which is dependent upon the simultaneous presence of microphones and loudspeakers within the same enclosure, instability due to acoustic feedback is the limiting factor in its performance. If the system is operated at the point of instability for a particular frequency, the effective reverberation time in the enclosure is infinite for that frequency. If the gain of the system is reduced by, say, 2 dB from the instability point, the reverberation time will be equal to 30 times* the average time taken for a signal received by the microphone to pass through the entire system and return to the position of the microphone in the studio. If this circulation time is too short the reverberation time obtainable will be short whereas if it is made too long the individual cycling periods can be heard as unpleasant flutters. The space for manoeuvre between these limits is extremely small even though both internal gain and cycling time may be adjusted independently. The transmission characteristic of the room between typical loudspeaker and microphone positions has fluctuations of the order of 10 dB between frequencies quite close to each other. This means that when the system is set to ensure stability at all frequencies, the average margin of stability for all frequencies even within a small band may be as much as 5 dB or more and the average reverberation time correspondingly lower.

Many attempts have been made in the past by experiments in Television Centre Studio 4 to achieve satisfactory adjustment and to formulate setting-up instructions. This has been extremely difficult, partly because it has not been possible to obtain the use of the studio except for very short periods, and partly because the apparatus bay is inconveniently situated with respect to the desk in the studio. It has usually been necessary to make experiments in rehearsal periods when little work of an experimental nature could be attempted.

A theoretical examination was made in 1962 as a result of which it was possible to improve the distribution of the loudspeakers among the reproducing head amplifiers and this was later realised

* The reverberation time is the time for a reduction of 60 dB in the reverberant sound; hence 30 circuits at 2 dB loss are required.

as a computer programme which was found to give a further improvement in performance. Details of this work are given in Appendix I. However, it was felt that despite all the experiments up to this point the system required considerable study before it could be expected to give effective and reliable performance, and permission was obtained to install the bay and 30 loudspeakers in the Large Viewing Room at Kingswood Warren so that a thorough investigation could be carried out. The equipment was delivered on June 4th, 1965.

2. INSPECTION OF THE EQUIPMENT

When the equipment was received at Kingswood Warren, the whole system was examined to ensure that it was operating to the best advantage. It was already known that the idler wheel on the tape was badly worn and unusable for programme purposes. A new one was obtained by Television Service and reached Kingswood Warren by June 30th enabling testing to start.

In the examination it was discovered that the internal feedback was derived from heads Nos. 4 and 5 instead of from head No. 5 alone as recommended on the basis of the theoretical examination and previous experiments. There was also a direct parallel connection from the microphone input to power amplifier No. 4 so that all loudspeakers connected to this amplifier carried both direct and delayed signal, and heads Nos. 3, 4 and 6 were connected to the wrong pre-amplifiers, thus invalidating all previous calculations. The connections between the heads and pre-amplifiers were difficult of access and, being soldered to a tag board, not easy to change round when required. Some connections were in rather poor condition and all were therefore cleaned and remade. As the performance of the system depends upon the distribution of the loudspeakers, it would be very desirable if the necessary permutations of heads and pre-amplifier could be made more easily. To this end, a miniature jackfield enabling the connections to be rearranged rapidly was finally fitted in place of the permanent connections.

Hum was excessive on the recording system even with all head screens and deck screen in place. There was also excessive noise, presumably from the power amplifiers, which we have not attempted to improve. During the tests, particularly the recordings described later, this hum and noise were found to be disturbing and consideration should be given to reducing it in any future equipment. The modification to the head connections produced an incidental reduction in hum pickup.

The head connections as found are shown in column 3 of the following table. Column 2 shows the head connections previously recommended and the final column the connections found best during the present work.

PRE-AMPLIFIER	HEAD CONNECTIONS		
	ORIGINALLY RECOMMENDED	AS FOUND	PROPOSED NEW CONNECTIONS
1	1, 2	1, 2, 3	1, 2
2	3, 4	6	3, 5
3 (and Feedback)	5	4, 5	4
4	6, 7, 8	7, 8	6, 7, 8

3. EXPERIMENTAL WORK

3.1. Internal Cycling Time

The internal feedback in the tape loop system can be taken from any of the reproducing heads and the cycling time may be varied from 30 ms up to 240 ms. If the internal cycling time is made too long there are very prominent flutters associated with transients in the sound. If, on the other hand, the cycling time is made too short, the flutter repetition rate is heard as a low-frequency colouration with its harmonics. If the feedback is taken from the first head, for instance, colourations are heard at all multiples of 33 c/s. Figs. 1(*a*) and 1(*b*) are pulsed glide displays[2] showing on (*a*) long isolated decays due to the harmonic colourations caused by too short a cycling time, and on (*b*) the result when the system is in an acceptable state of adjustment. The best compromise was found to be feedback from the 4th head; head 5, which was originally recommended in 1962, gives more noticeable flutters while head 6 was unacceptable. Figs. 2(*a*), 2(*b*) and 2(*c*) show short pulse displays with feedback from the 4th, 5th and 6th heads respectively. To obtain these oscillograms a logarithmic amplifier was inserted at the input to the oscilloscope so that a normal exponential decay appears as a wedge-shaped envelope. The increasing flutter components in 2(*b*) and 2(*c*) may be seen as regular fluctuations in the envelope and are marked 'x'.

3.2. Equalisation

It has already been noted that inefficient performance is caused by an irregular transmission characteristic for the system, since instability may occur at a particular frequency while sounds of all other frequencies have a large stability margin. Equalisation over the whole frequency range is therefore essential to ensure the greatest effectiveness even though it is clear that it is not possible to remove the close-spaced fluctuations within any small band of frequencies.

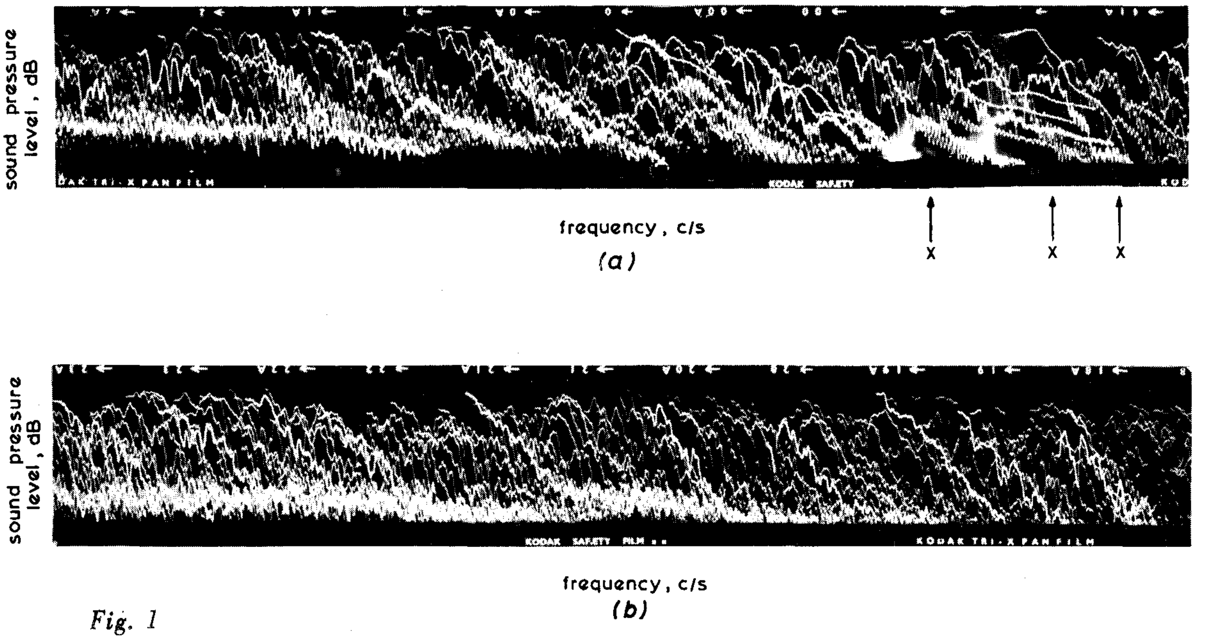

Fig. 1

(*a*) Pulsed glide of system with feedback from 1st replay head. (note long decays at x)
(*b*) Pulsed glide of system with feedback from 5th replay head.

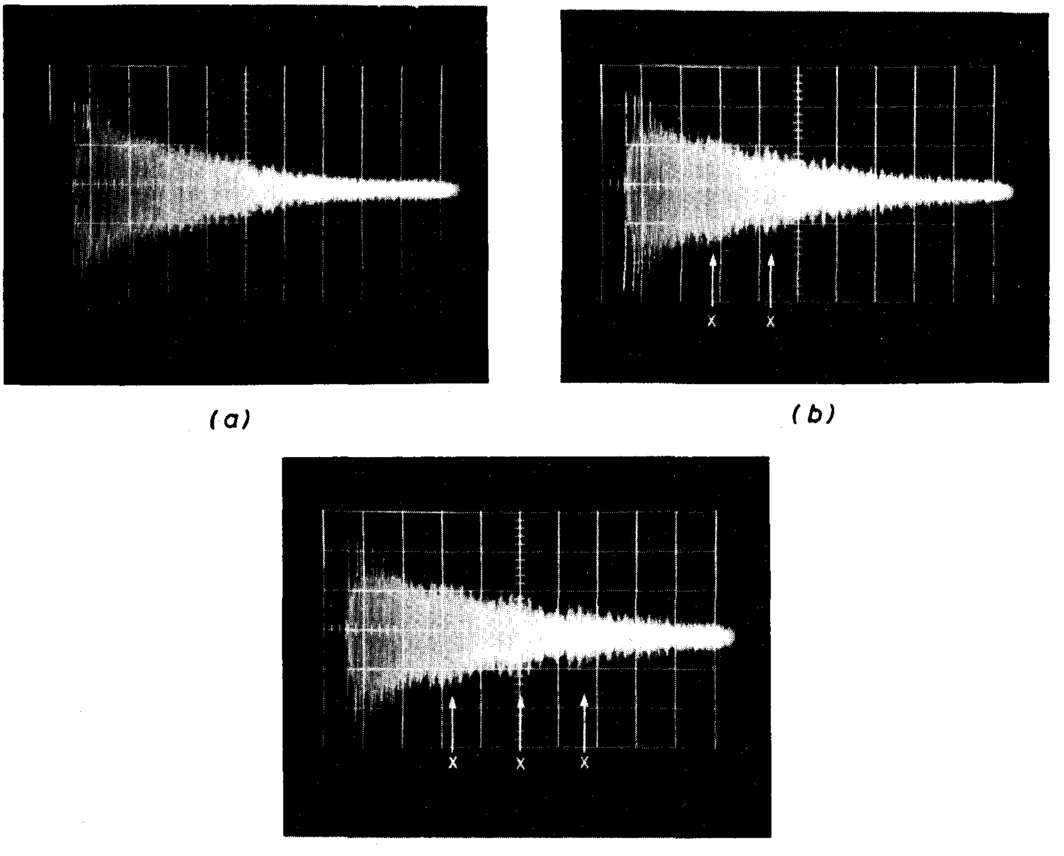

Fig. 2

(*a*) Short pulse display of ambiophony system, feedback from 4th head
(*b*) Short pulse display showing fluctuations at x due to feedback from 5th head
(*c*) Short pulse display showing fluctuations at x due to feedback from 6th head

The first necessity in setting up the system is therefore to inject "pink" noise (random noise with constant power per octave bandwidth) into the recording amplifier of the equipment through a variable filter element and to measure the sound pressure level in the studio in a succession of octave bands by means of a microphone amplifier and any suitable detector such as a peak programme meter. The transmission characteristic of the whole system is then adjusted by means of the variable filter element until all bands up to that centred on 1·4 kc/s give equal sound pressure levels, after which the gain should fall off by 3 dB/octave. This ensures adequate effect in the middle frequency region where reverberation is most important without introducing instability in the high frequency region, otherwise found to be virtually unavoidable.

A final adjustment is made by measuring the reverberation time of the studio at one-quarter octave intervals, with ambiophony switched on and using the equalisation filter to remove any prominent resonances shown by peaks in the reverberation characteristic.

It should not be necessary to alter the equalisation of the system once it has been installed and adjusted, and since the performance is very sensitive to equalisation, it is recommended that the variable filter element be replaced by a fixed equaliser as soon as the best adjustment has been found. Fig. 3 shows the reverberation time characteristics obtained in the Viewing Room (*a*) without ambiophony and (*b*) and (*c*) with the two successive stages of equalisation.

3.3. Loudspeakers

The loudspeakers installed in the Television Centre Studios are of two types having different fundamental resonances. The object of this was to give as smooth as possible a frequency characteristic in the bass without the use of expensive loudspeakers. It appeared at one time that this might have been a mistake and that improved stability could be gained by using a single type so that all loudspeakers would require identical equalisation. Experiments in the Viewing Room in which all loudspeakers of one type were disconnected showed, however, that the use of mixed types is not inferior. It was also found preferable to add a number (at least one per power amplifier) of loudspeakers with a good performance at low frequencies. This was not necessary in the interests of maintaining a high reverberation time but it resulted in a marked improvement in the subjective quality of the reverberation.

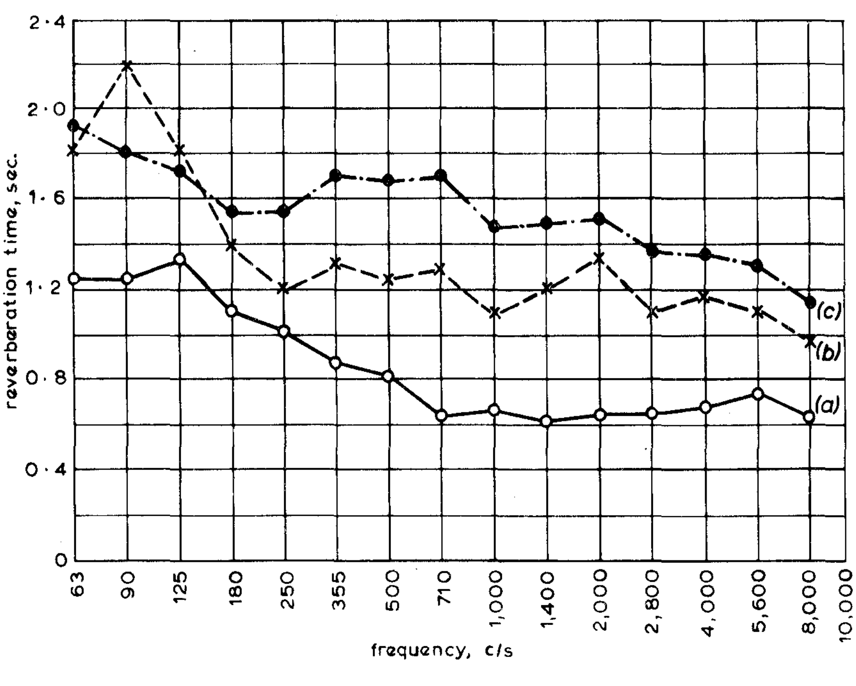

Fig. 3 - Effect of equalisation of ambiophony on reverberation time

(*a*) Ambiophony off
(*b*) Ambiophony on, speakers equalised on 'pink' noise
(*c*) Ambiophony on, speakers equalised on 'pink' noise, equalisation further adjusted to reduce isolated long tails on decay curves

3.4. "Direct Loudspeakers"

In order to obtain the parabolic rise with time in the echo density required to simulate natural reverberation,* the use of loudspeakers without delay would appear to be indicated and in the early experiments in Television Centre some of the loudspeakers were connected directly to the input to the recording head. However, there is a disadvantage in the use of undelayed loudspeakers since they give the effect of a reflecting wall halfway between the source and the studio wall and therefore make the studio sound smaller. Although no experiments in the Large Viewing Room were specifically planned to find the effect of the undelayed loudspeakers, a considerable improvement was noticed when the direct connection to amplifier 4 mentioned above, was discovered and removed.

The reason for this result is probably that early echoes at the required density are provided by the floor which is a reflecting surface in both the Viewing Room and a television studio. The use of undelayed loudspeaker connections is therefore unnecessary in any television studio to which the application of ambiophony is envisaged.

3.5. Operating Distance of the Ambiophony Microphone

All experiments demonstrated that an ambiophony microphone close to the source of sound is absolutely necessary for a worthwhile effect. Six feet (2 m) should be considered the maximum practicable working distance of the nearest ambiophony microphone from the source of sound, and if increased distances are used the ambiophony system will produce rapidly diminishing reverberant sound intensities unless it is worked so near to the limit of stability at some frequencies that the result is highly coloured. Fig. 4 shows reverberation curves corresponding to three different ambiophony microphone distances with the system set to give the maximum output without serious colouration.

3.6. Directional Microphone

Experiments were carried out with directional microphones (Electrovoice 642) both with the pulse and reverberation equipment and also with music recordings. This microphone is highly directional at middle and high frequencies and its directivity diagram becomes a cardioid at low frequencies. It was found possible to use it at about twice the distance of an omnidirectional microphone and obtain a comparable reverberation time and intensity of reverberant sound in the studio. The music recordings, whilst confirming this, were judged inferior generally in quality to those obtained with the omnidirectional microphone, probably due to the frequency discrimination in the directional properties. It was found, however, that very long reverberation times could be obtained at low frequencies by the use of omnidirectional microphones.

* See Appendix I

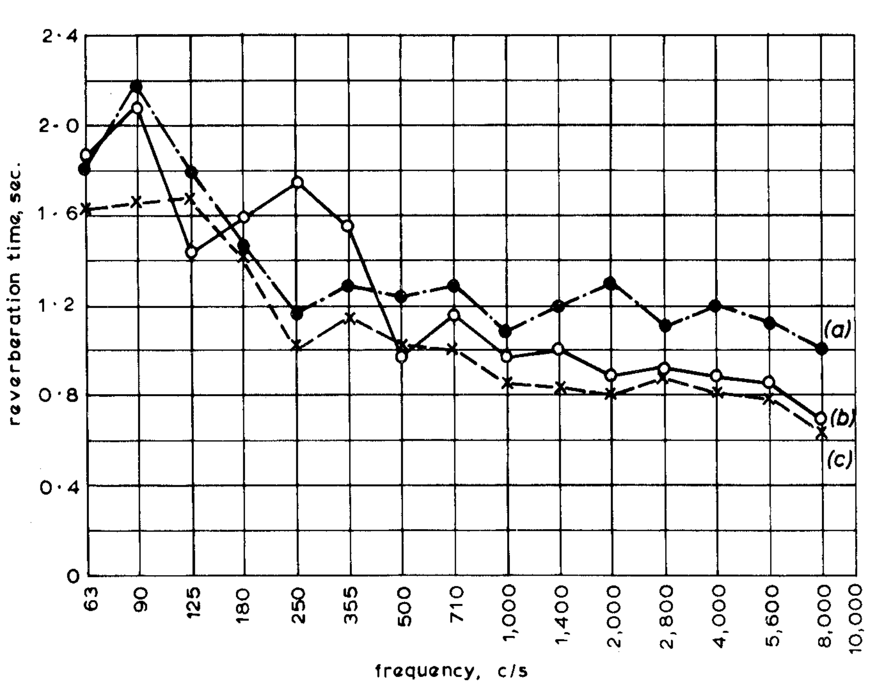

Fig. 4 - Variation of reverberation time with distance from sound source to ambiophony microphone
(*a*) 2 ft (600 mm) (*b*) 6 ft (2 m) (*c*) 27 ft (8 m)

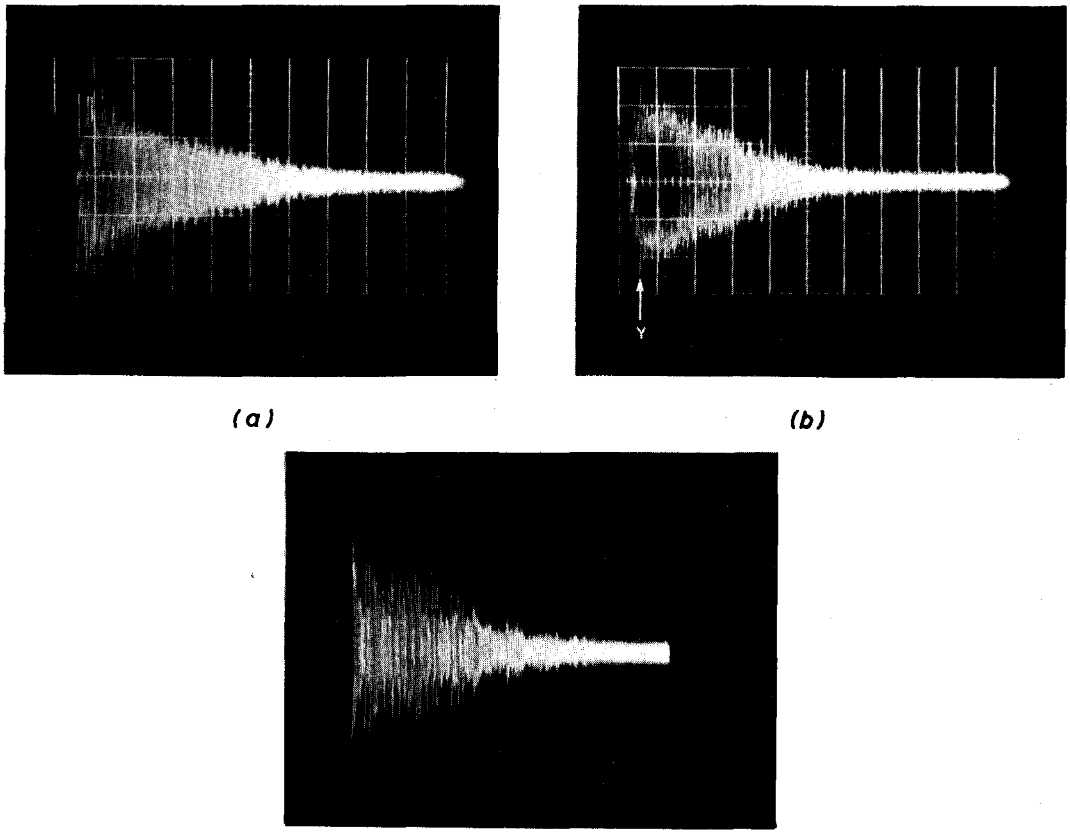

Fig. 5

(*a*) Short pulse oscillogram with system in good adjustment
(*b*) Short pulse oscillogram with programme microphone nearer to nearest loudspeakers than to original source of sound. (Note strong peak at Y)
(*c*) Short pulse oscillogram before carrying out the present work. Note roughness and irregularities compared with 5(*a*)

3.7. Interconnection of Heads, Amplifiers and Loudspeakers

There was insufficient time to investigate thoroughly the influence of the interconnections between loudspeakers, amplifiers and heads except with regard to the internal feedback arrangement mentioned in Section 3.1. To carry out this work properly it would have been necessary to re-write the computer programme to make it suitable for 30 loudspeakers only. As a compromise it was decided to use the original computer programme including in the data the actual co-ordinates of the 30 loudspeakers and hypothetical positions for the other loudspeakers too far away to have any effect on the figure of merit.* When new connections had been assigned and made there was a measurable improvement in the system.

Having regard to the adventitious direct connection to amplifier 4, and the incorrect connections between amplifiers 1, 2, 3 and heads 3, 4 and 6, a new figure of merit was computed for the most recent arrangement in T.C.4 and it was found to be grossly inferior to that obtained for the connections specified at the time.

* i.e. at such a distance that the first echo occurs after the critical time (see Appendix I).

Although no systematic experiments were made to this end during the period when the equipment was at Kingswood, there were two incidental experiments which produced an improvement in the calculated figure of merit and these also produced quite marked improvements in the subjective performance.

3.8. Pulse Echo Distribution

Work on the echo time distributions was guided generally by photographs of the oscillograph traces displaying the amplitudes and the arrival times of echoes of a short pulse of high frequency tone radiated in the room. Figs. 5(*a*), 5(*b*) and 5(*c*) show three such oscillograms. Fig. 5(*a*) shows the system in an acceptable state of adjustment. Following a fairly rapid initial rise there is an exponential decay down to noise level. The direct pulse will be seen as an isolated spike at the beginning of the decay. Fig. 5(*b*) shows the result when the distance from the sound source to the programme microphone is greater than the distance from the programme microphone to the nearest non-delayed loudspeakers. The slower rise time and the occurrence of a strong direct peak after an interval of 35 ms from the start of the trace can easily be seen at ‘Y’. Fig. 5(*c*)

shows an oscillogram obtained in Studio 4 Television Centre before the system had been adjusted. The irregular decays and pulse groupings which led to instability are evident in the photograph which should be compared with Fig. 5(*a*).

3.9. Use of P.A. Stabiliser

The Public Address Stabiliser developed by Research Department from the ideas of Schroeder[3] was tried as a means of increasing the stability of the system and thereby increasing the total effective reverberation. This equipment is designed to introduce an adjustable frequency change of up to 5 c/s between the input and output terminals. It was inserted between the ambiophony microphone and the bay, therefore enabling an increase of about 4 to 6 dB to be obtained in the stability margin for any given setting of the ambiophony system. However, the results on music were unacceptable owing to the introduction of beats between the live sound and the ambiophony sound; these were particularly noticeable on sounds of low frequencies, such as the pedal notes of the electronic organ in the Viewing Room.

4. DISCUSSION OF RESULTS

4.1. Microphone Working Distance

The results which show the benefit of close ambiophony microphones and the inadvisability of working at distances much greater than 6 ft (2 m) have important practical implications. For solo voices or instruments it is easy to achieve the optimum working distance, but on large orchestral ensembles new problems present themselves. Placing the ambiophony microphones so that they are out of camera shot will reduce the ratio of direct to reverberant sound at the input to the ambiophony equipment, thus increasing the colourations introduced by the characteristics of the system and also reducing the stability margin. A significant increase in liveness in the studio without the introduction of severe colourations can be obtained only if the ambiophony microphones are situated close to the sound source. Producers of orchestral music programmes in television studios will have to face the necessity of using ambiophony microphones in shot, in order to bring them within a practical working distance for the ambiophony system. In the past this suggestion has met with considerable opposition on visual grounds, and there seems a case therefore for developing or purchasing less obtrusive microphones specially for this purpose. It should be mentioned here that the limitation imposed on microphone distance was mentioned in the original paper on stereo-reverberation by Vermeulen.[1]

4.2. Application to Other Studios

It is proposed in the near future to make ambiophony available in Television Centre Studio 1 as well as in the smaller studios. As this studio is considerably larger than the others, special attention will have to be given to the installation and setting up. It is very probable that the delay of the first head will have to be increased to 45 ms in place of 30 ms since the dimensions of the studio are larger and the walls consequently farther away from the working area.

The present investigation also confirmed that it is necessary to set up the equipment by a series of tests in the studio in which it is to be used, and

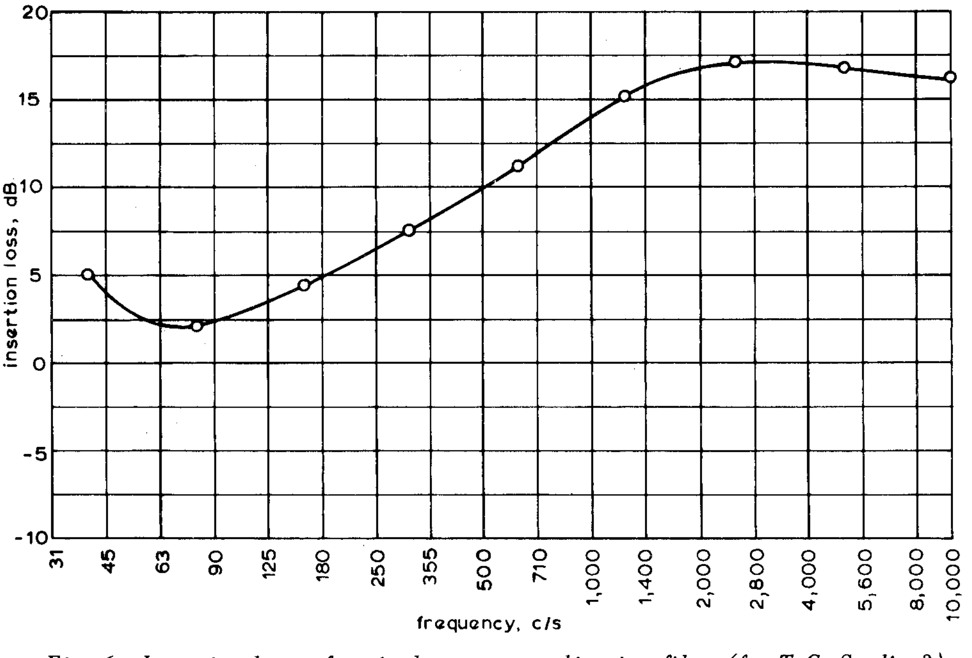

Fig. 6 - Insertion loss of typical preset equalisation filter (for T.C. Studio 3)

that reliable results can be obtained only if this is done. The existing arrangements at Television Centre allow for one bay of equipment only, which is connected to loudspeakers and microphones in four studios by means of tie-lines. In these circumstances we can see no way of setting up the equipment to the best advantage in one of the remote studios; it is difficult enough to do so in Studio 4 with the bay of equipment located in the Apparatus Room on the far side of the Sound Control Cubicle and would be worse with longer lines of communication. If it is considered inadvisable at this stage to equip each of the studios with a complete equipment bay, a cheaper method which has some advantages might be considered. The three "Satellite" studios Nos. 1, 2 and 3 would be served by a trolley bay carrying only the four power amplifiers. This bay would be attached by a multi-core flexible cable to a socket in the Sound Supervisor's desk and would thus be in the best position for adjustment while listening. Pre-set zero-gain equalisers would be mounted in each studio; a typical equaliser characteristic (for T.C.3) is shown in Fig. 6. The existing tape delay unit would remain in Studio 4 Sound Control Room and be plugged in to the lines leading to the other studios.

5. CONCLUSIONS

The following conclusions arise from the experimental work of the last two months together with experience gained since ambiophony was first installed in the television studios.

1. The microphone or microphones connected to the ambiophony system must be as close as practicable to the sources of sound for the system to be effective and in no case substantially farther than 6 ft (2 m). This may entail the use of ambiophony microphones in camera shot and if this is considered visually unacceptable, less obtrusive microphones will have to be obtained. There is no such restriction on the distance of the programme microphone.

2. Somewhat larger working distances can be obtained by the use of highly directional microphones. Further experiment is required on this since increased working distance and effectiveness of the system are partly offset by the deterioration in tonal quality.

3. The ambiophony equipment bay should be as close as possible to the sound control position of the studio served by it since individual adjustment of the equipment is required for each studio and final adjustment can be made only with the use of the monitoring loudspeaker. This implies that ideally there should be one bay for every studio equipped for ambiophony. A less expensive compromise is suggested.

4. The equipment must be equalised to obtain a suitable frequency characteristic and when this has been done the equalisation should be fixed. A separate equaliser should therefore be made for every studio and its associated equipment.

5. The method of allocating interconnections between loudspeakers and amplifiers by computer appears to be successful and there is good correlation between the figures of merit obtained by calculation and the subjective performance.

6. The use of the P.A. Stabiliser developed for sound reinforcement systems is not recommended since the increased stability is offset by tonal degradation.

7. The inclusion of a relatively small number of wide-range loudspeakers gives marked improvement in tonal quality.

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APPENDIX I

Theoretical Basis for the Method of Allocation of Connections between Loudspeakers, Amplifiers and Heads

In any studio the build up of sound is due to reflections from floor, walls and ceiling; the earliest reflections are from the floor. The only hard reflecting surface in a television studio is the floor, for the walls and ceiling are covered with material of high absorption coefficient.

Suppose that a live room is excited by a short impulse, then we first receive an impulse from the direct sound, followed by early reflections from the floor then from the walls and ceiling. Gradually the echo density increases until after a critical time t_c depending on the volume of the room and the pulse length, the echoes begin to overlap and are no longer individually distinguishable.

The number n of echoes per unit time is related to the elapsed time t by:[4]

$$n = \frac{4\pi c^3 t^2}{V} \tag{1}$$

where V is the volume of the room and c is the velocity of sound.

If Δt is the pulse width, the maximum value of n without overlapping of echoes is $1/\Delta t$. Hence, by substitution of this value of n in equation (1) we get

$$t_c = 5.10^{-5} \sqrt{V/\Delta t} \tag{2}$$

where Δt is in seconds and V is in m^3. For T.C. Studios 3 and 4 which have volumes of 350,000 ft^3 (10,000 m^3) the critical time is 158 m.sec for a 1 m.sec pulse.

Thus when ambiophony is used in dead studios to simulate live room acoustics the system must be adjusted to fulfil the requirements of equations (1) and (2).

An impression of the dimensions of a room is conveyed by the time taken by early reflections from the walls to return to the listener. Thus if the room walls are highly absorbent and loudspeakers are placed on the walls in an attempt to simulate reflecting walls, the signal to the loudspeaker must not arrive at the loudspeaker before the arrival of the direct airborne sound from the sound source. If this is not so, then the dimensions of the room appear to be decreased.

The first attempt to distribute the loudspeakers among the several delays by the use of echo-spacing calculations was made in 1962. The object of the calculations was to examine the time distribution of arrival at the ambiophony microphone of all the echoes of a pulse injected into the system, and to eliminate long gaps or dense groupings of echoes.

An arbitrary but typical microphone position was chosen and the distance of every loudspeaker from this point was calculated by solid geometry from its co-ordinates. These distances were converted into times by dividing by 340 m/s (the speed of sound in air at 20°C) and were added to the magnetic tape delay times so as to give the total time of transit through the system.

On plotting these times on a time-scale it was found that there was considerable bunching of echoes at certain times. The delays of groups of loudspeakers were then rearranged until a satisfactory diagram was obtained.

This process, though it effected an immediate improvement, did not take into account the additional necessity for the correct law of build-up of echoes; it also involved long periods of calculation by desk calculator.

With these factors in mind a computer programme has been written in which connections between amplifiers and loudspeakers are selected at random in an attempt to synthesize the required parabolic echo build-up. In addition marked echo grouping after the critical time has elapsed is avoided. A figure of merit based upon deviation from the parabolic build-up law is computed for the selected set of connections. The selection process is continually repeated, ideally until a sufficiently steady maximum figure of merit is attained. In practice the programme carries out about 200 selections, tabulates the selection with the highest figure of merit, and prints out the pulse echo time distribution.

Any particular set of connections is strictly applicable for only one position of the programme microphone but will be satisfactory over a useful area around that position. In practice, therefore, it should be unnecessary to compute and change connections for every orchestral programme although this might be necessary in the unlikely event of a major departure from the normal layout.

APPENDIX II

Procedure for Adjustment of the Ambiophony Equipment

1. Set all amplifier tone controls to 'flat' and switch on the equipment.

2. Inject 'pink' noise at the input to the ambiophony bay through a variable equalisation unit.

3. With the programme microphone in a typical location in the studio, measure the sound pressure level in each octave band. Adjust the response of the equalisation unit to give equal sound pressure levels in each octave band up to a mid-band frequency of 1·4 kc/s, thereafter falling by 2 to 3 dB per octave.*

4. Restore the normal input to the ambiophony system with the "Rev. Time" control set to minimum; switch on the power amplifiers one at a time, increasing the gain of each amplifier in succession to a point just short of instability. If separate low-frequency loudspeakers are used, bring these into operation in succession, again adjusting the gains to a point just short of instability.

5. The longest usable reverberation time is determined by the presence of flutters in the decays caused by too high a setting of the "Rev. Time" control. The "Rev. Time" control should be set as high as is possible without introducing flutters.

* For optimum performance the equalisation should be finally adjusted during reverberation time measurements to give a flat reverberation characteristic at frequencies up to 1·4 kc/s.

CHD

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